Manager Manual

IP Phone

Version 2.3.1

Startcom Information&Technology Co.,Ltd

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1 Welcome

IPT1000H are innovative gateways that offer a rich set of functionality and superb voice quality. They are fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

2 Installation

2.1 Package List

The IP Phone package contains:

1) One IPT1000H IP Phone

2) One universal power adapter

3) One Straight Ethernet cable

2.2 Safety Compliances

The IP Phone should only be operated with the universal power adapter provided with the package. Damages to the IP Phone caused by using other unsupported power adapters would not be covered by the manufacturer’s warranty.

3 Product Overview

IP Phone is a next generation IP network facility based on industry open standard SIP (Session Initiation Protocol). Built on innovative technology,

IPT1000H features market leading superb sound quality and rich functionalities.

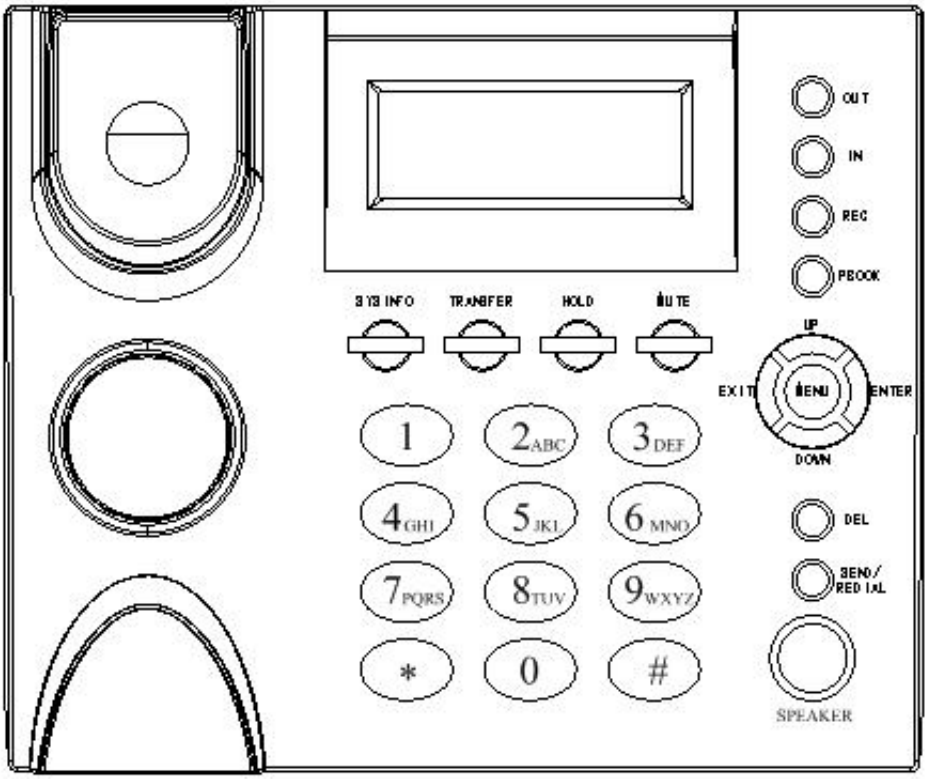


Fig 3.1 IPT1000H Front Illustration

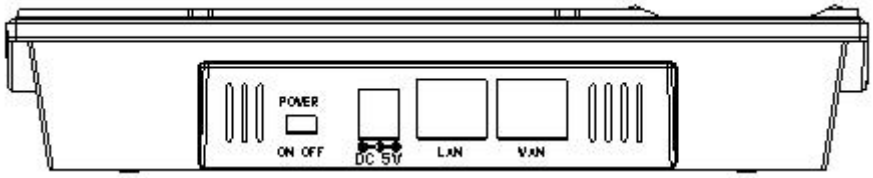


Fig 3.2 IPT1000H interface sketch map

- The WAN connects: With Server or HUB to connect with each other.
- The LAN connects: With the calculator or the other network equipments conjunction.

3.1 key Features

- Support two models: Bridge and Router(NAT&NAPT)

- Network Protocols: TCP/UDP/IP, ICMP, HTTP, DHCP Client（WAN Interface）, DHCP Server（LAN Interface）, DNS Client, DNS Relay, SNTP, PPPoE, FTP, TFTP

- VoIP Protocols: SIP (RFC3261)

- Voice Codecs: G.711A/U、G.723-r63、G.723-r53、G.729

- Redundancy SIP server: Can auto swap address between two servers address

- NAT transversal: Support STUN client. Can modify SIP register port, HTTP server port, Telnet server port and RTP port

- Support two SIP servers synchronously: Can register two different SIP servers, and can make a call by either proxy

- Support standard voice features such as numeric Caller ID Display, Call Waiting, Hold, Transfer, Do-Not-disturb, Call Forward, in-band and out-of-band DTMF, Hotline (off hook autodial), ban outgoing

- Support standard encryption and authentication (DIGEST using MD5, MD5-sess)

- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)

- Provide easy configuration thru manual operation (Web interface or Telnet) or automated centralized configuration file via TFTP or HTTP.

- Support firmware upgrade via TFTP/FTP and HTTP

3.2 Hardware Specification

The table below describes the hardware specification.

Item		Specification
Main chip		Infineon PSB21553
SDRAM		8 MB
Flash memory		2 MB
Power Adapter	Input	100-240V AC
	Output	5V DC 1A
Port	WAN	1 10/100Base T RJ-45
	LAN	1 10/100Base T RJ-45
Power Consumption		1.4W/1.8W
Operating Temperature		0～40℃
Relative Humidity		10～90%
Volume		200mm*185mm*70mm

3.3 Hardware Specification

1) Connect handset to base: insert handset cord into handset cord jack at the left side of the base.
2) Connect IP phone to internet: plug the Rj-45 Ethernet cable into the Ethernet Jack. Plug the other end of the cable into HUB or switch.
3) Connect IP phone to internet: plug the Rj-45 Ethernet cable into the Ethernet Jack (RJ45). Plug the other end of the cable into

HUB switch.

4) Connect PC to internet : plug the RJ-45 Ethernet cable into the Ethernet Jack (PC). Plug the other end of the cable into PC machine.

5) Power on IP phone: plug the power cord adapter into the Power Jack. Then plug the other end of the power cord adapter into the appropriate wall outlet.

With the installation done, users can chose the telephone keyboard or HTTP orders according the preference of each, and set the respective parameters. Details may refer to the related chapters in the manual.

4 Configuration with Phone

We have provide some function keys to config IPT1000H_v2.3. We can connect one analog phone to “Phone” interface.

4.1 Function Keys:

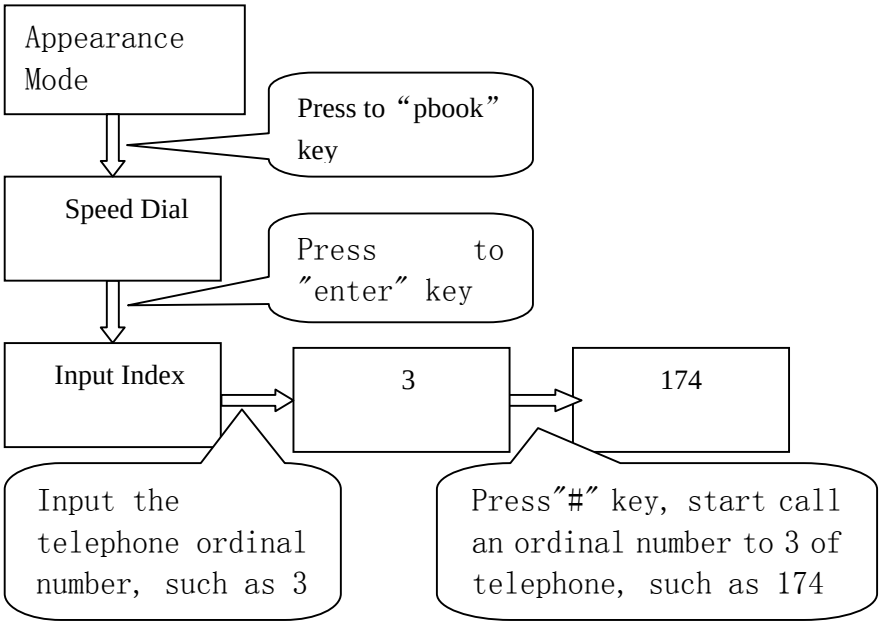
Key	Operation	Function/display
SYS INFO	Alert Mode	Show phone number, WAN IP address and gateway IP address
ENTER	Configure Mode	Enter next class menu and confirm function
EXIT	Configure Mode	EXIT
UP	Talking Mode	Raise a volume(biggest install to 9)

	Configure Mode	Install the page choice(heading up to turn over page)
DOWN	Talking Mode	Lower a volume(the least install to 1)
	Configure Mode	Install the page choice(get down to turn over page)
DEL	Calling Mode	Delete current number
	Configure Mode	Delete current value
MUTE	Talking Mode	MUTE
OUT	Alert Mode	Enter the menu of out call
IN	Alert Mode	Enter the menu of in call
REC	Alert Mode	Enter the menu of voice record
PBOOK	Alert Mode	Enter the menu of phone book
SPEAKER	Talking Mode	SPEAKER
0—9	Calling Mode	Common numeral 0-9
	Configure Mode	Fast press to show numeral and once, fast press the second to descend(perhaps the n descend) the manifestation English character list perhaps other character lists
*	Talking Mode	"call holding" appearance descends to press this key to carry out three way call.
	Configure Mode	Equal to 192.168.10.202 medium decimal points
#	Talking Mode	Can conduct and actions a number stir or equal to accept number be over marking
HOLD	Talking Mode	Call holding
TRANSFER	Talking Mode	Call transfer
SEND/REDIAL	Talking Mode	Send value or Heavy stir the last time call sign number to carry on call sign

4.2 In common use key elucidation

- Grow to press numerical one key 3 second, the IP Phone links network the mode cut over to the Static Mode.
- Grow to press numerical two key 3 second, the IP Phone links network the mode cut over to the DHCP Mode
- Grow to press numerical three key 3 second, the IP Phone links network the mode cut over to the PPPoE Mode

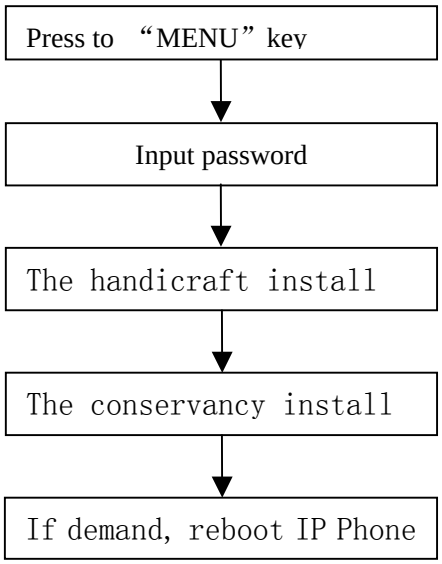
4.3 Stir number operation process quickly


Appearance Mode
Press to "pbook" key
Speed Dial
Press to "enter" key
Input Index
3
174
Input the telephone ordinal number, such as 3
Press"#" key, start call an ordinal number to 3 of telephone, such as 174

4.3 The IP Phone keyboard installs menu

Pass the IP Phone front-panel up of the keyboard carry on some install basically. If the WEB method can't register or the IP conflict, can install some parameters through a keyboard, then be carry on with WEB to install in detail.

The keyboard installs process



※ Note: If have no importation password, then enter a search appearance, can't modify to install.

4.3.1 The NETWORK installs menu

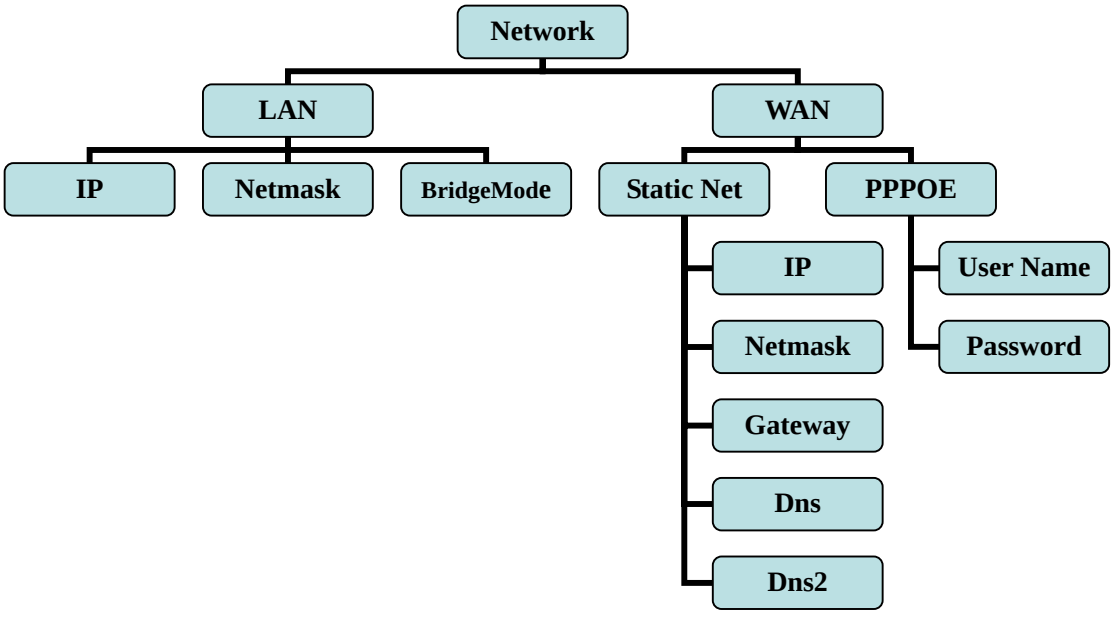


Diagram 4.3.1.1:The Network installs menu

IP Phone factory of the tacit approval establish: WAN IP is 192.168.1.179, LAN IP is 1921.168.10.1.

What to need to be notice : Recognize tacitly IP Phone while change

an IP of the bridge mode be in use, so WAN IP and LAN IP of the IP Phone can't belong to same net segment, wanting for example IP to change the LAN 192.168.1.1, have to first promise the WAN of IP Phone the IP isn't 192.168.1.X.

With the keyboard install the IP address of the WAN for example, other parameter handicrafts install to manage together.

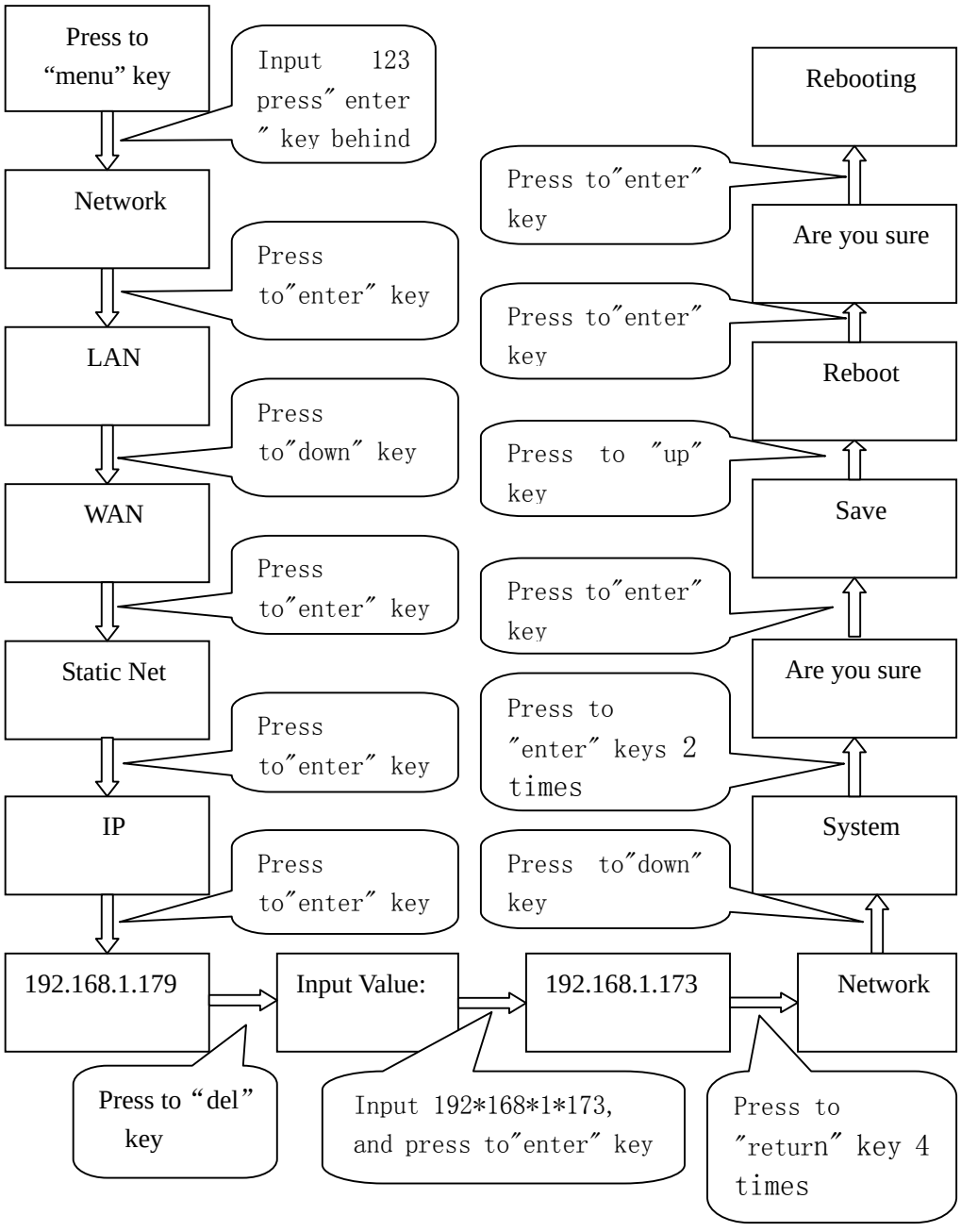


Figure 4.3.1.2: The handicraft installs the IP address of the WAN

4.3.2 The DSP installs menu

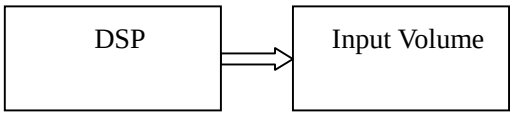


Diagram 4.3.2.1：**DSP** installs menu

The DSP installs only an item in the menu, be install the words tube volume size, scope is 1-9.

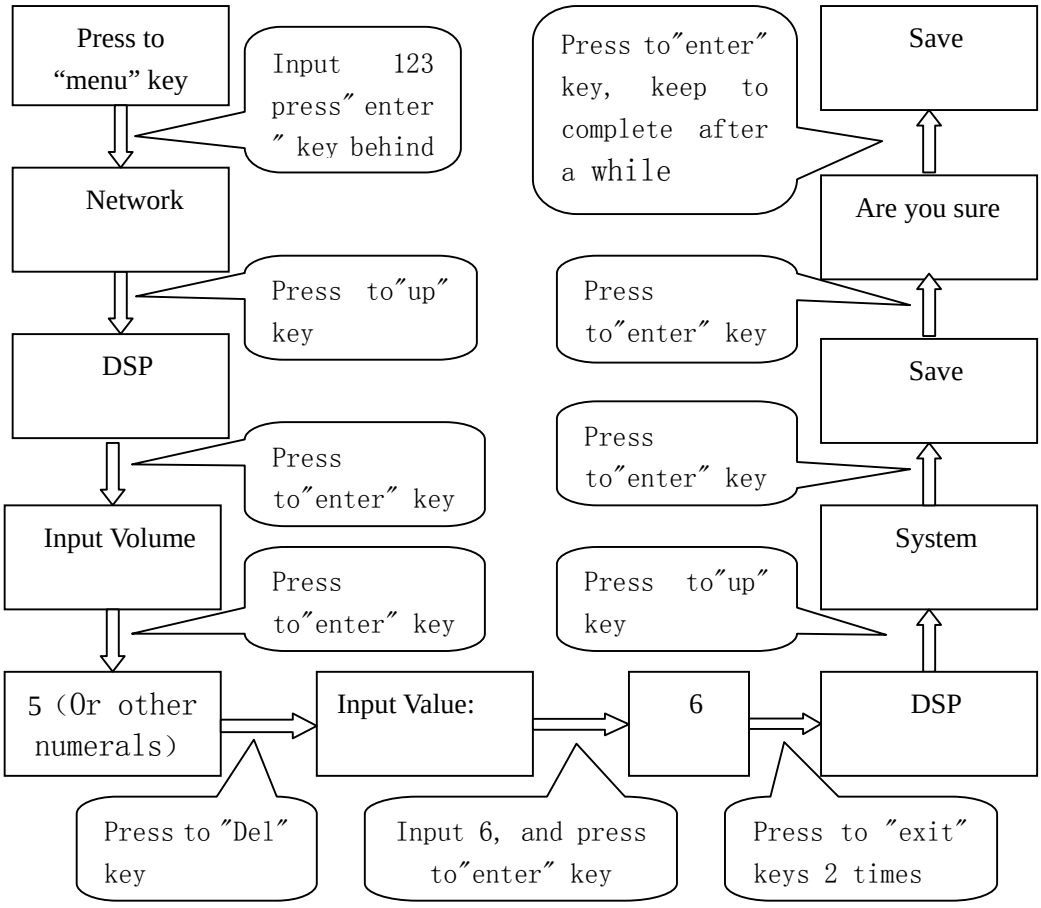


Diagram 4.3.2.1：Install words tube volume size

4.3.3 The SYSTEM installs menu

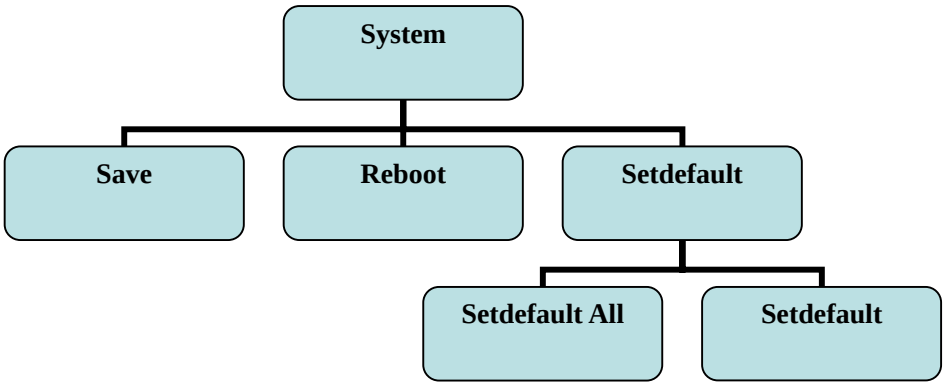
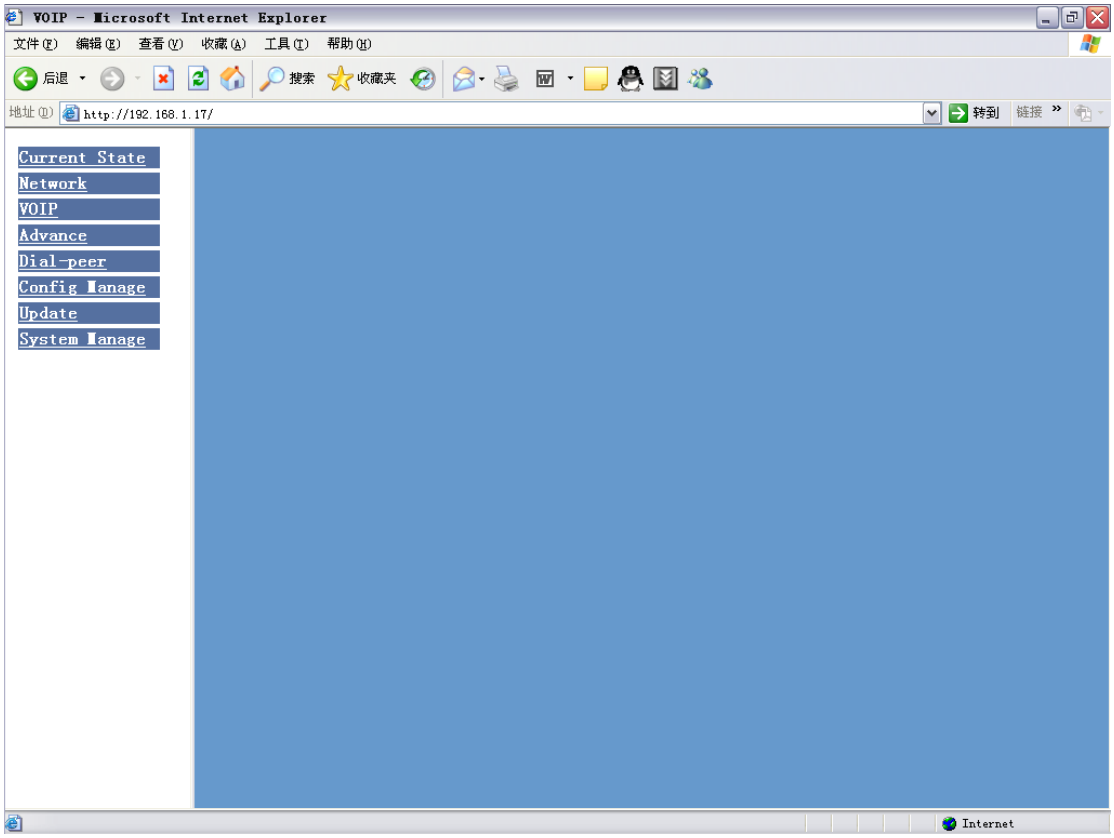


Diagram 4.3.3: The SYSTEM installs menu

5 Configuration with WEB

The IP Gateway Web Configuration Menu can be accessed by the following URL: *http://Gateway-IP-Address*. The default LAN IP address is "**192.168.10.1**" and WAN IP address is "**192.168.1.179**". otherwise the web will show that no server has been found. It will be shown as follows:



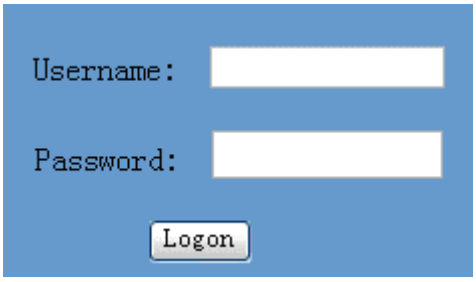
5.1 User verification

User should login before configuration dialogue machine.

Guest account: the default username and password are all “guest”, user can have a browse of system and configure the most of item of system exclude SIP.

Administrator account: the default username and password are all “admin”, this user can configure the system.

Note: after inputting username and password, user press “Enter” directly to enter the page.



5.2 Current state

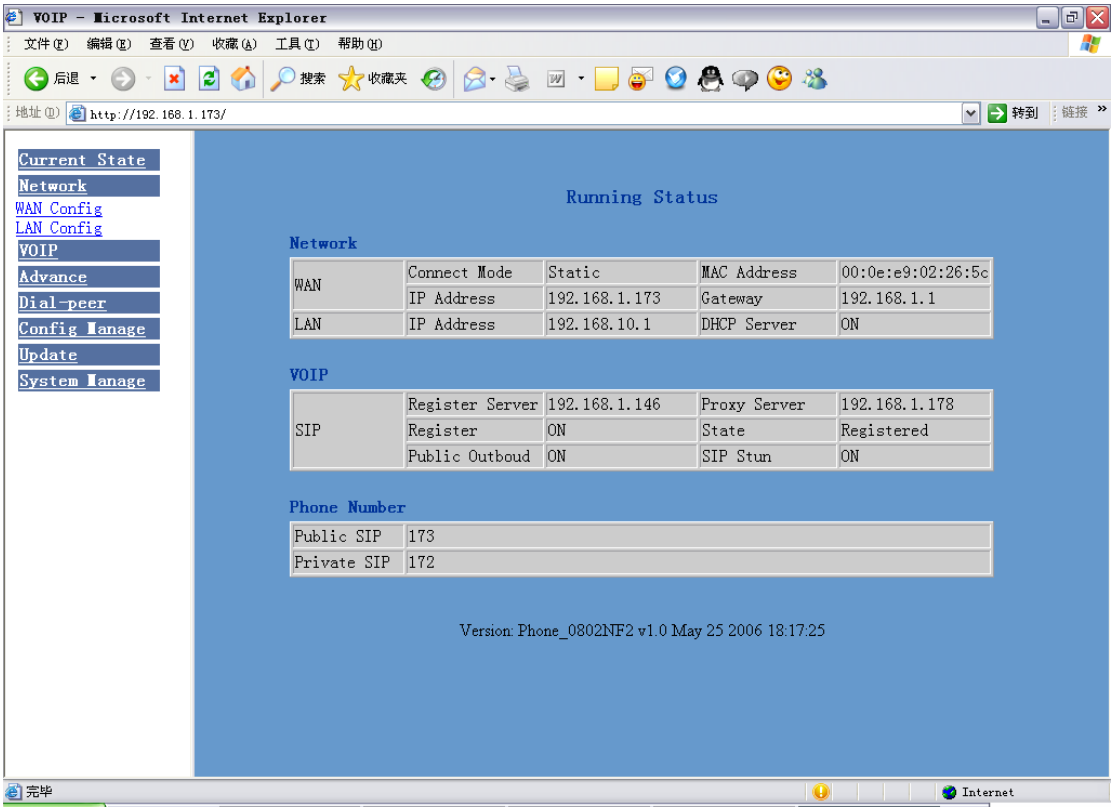
On this page user can gather information of each commonly-used parameter of the IP Phone, it is shown as the following figure: the network section shows the current WAN, LAN configurations of the IP Phone: including gaining way of WAN IP and IP address（static state, DHCP, PPPoE), MAC address, WAN IP address of the IP Phone, LAN IP address of the IP Phone, opening state of LAN DHCP server.

The VoIP section shows the current protocol in use, and server parameters in use of each protocol: including Register server IP of SIP, proxy server IP address, whether enables register, whether has registered on register server, whether enables outbound proxy, whether enables STUN server.

The Phone Number section shows corresponding phone number of each

protocol.

The version number and date of issue have been shown at the end of the page.



5.3 Network configuration

5.3.1 Wide area network（WAN）

User can view the current network IP linking mode of the system on this page.

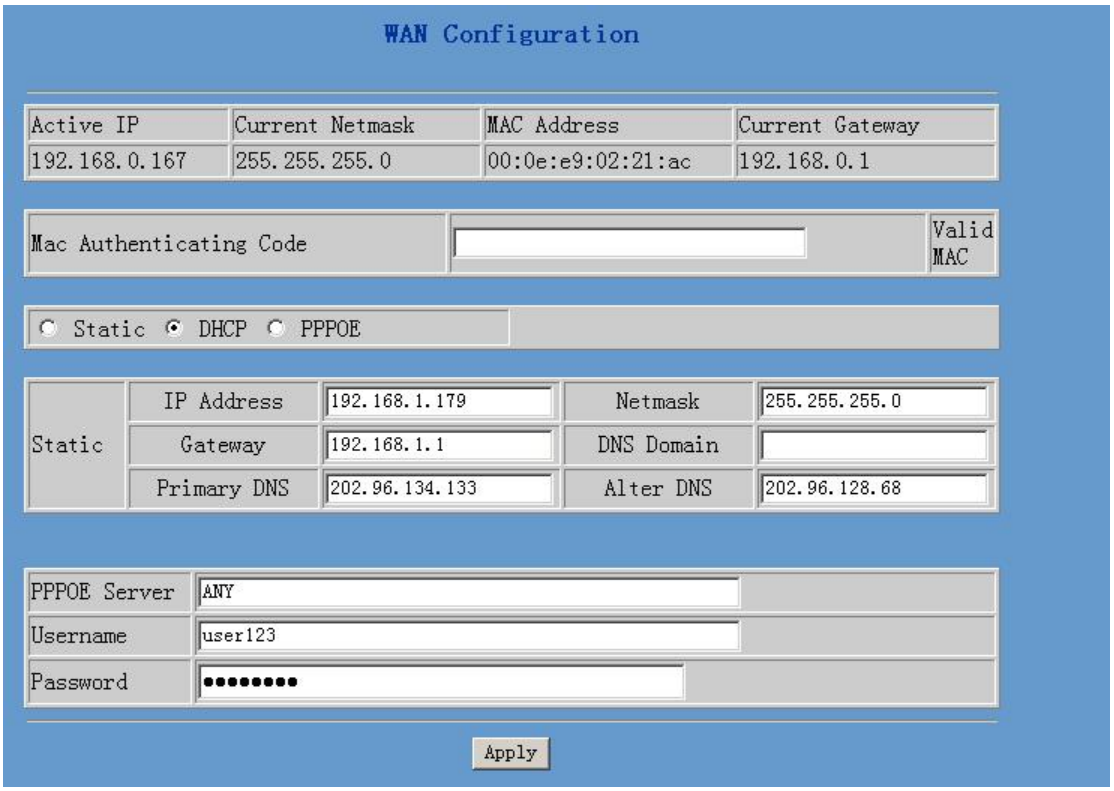
User will be authorized to set the network IP, Gateway and DNS if the system adopts the static linking mode.

If the system selects DHCP service in the network which is using DHCP service, IP address will be gained dynamically.

If the system selects PPPOE service in the network which is using the

PPPOE service, then the IP address will be gained by the set PPPOE ISP internet and password of the account.

Note: if IP address has been modified, the web page will no longer respond owing to the modification, so new IP address should be input in the address field now.



Configuration Explanation:

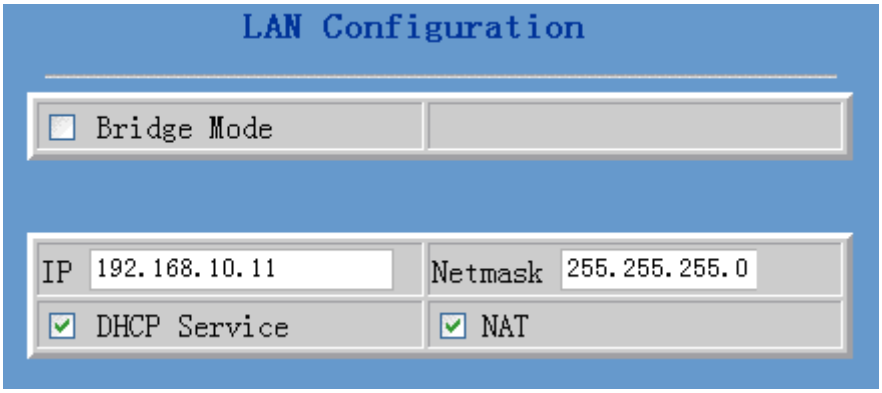
First Table: Active IP / Current Netmask/ MAC Address/ Current Gateway	Current IP, subnet mask, MAC address and current gateway IP;
Mac Authenticating Code	To confirm if the MAC Address is legal. If legal, it displays Valid MAC, the device works normally; if illegal, the IP Phone has no dialling tone and can not

Code	make a call, but other functions still work. When illegal, ask the device provider to get MAC check digit, then submit and restart, the IP Phone will be recovered.
Static/DHCP/ PPPOE	Select acquisition way of IP for WAN; This is single option; Configure static IP parameter for WAN;
Static Parameter Configuration:	
IP Address	Configure static IP address;
Netmask	Configure subnet mask;
Gateway	Configure IP address of the gateway;
DNS Domain	Configure “dns domain” suffix; if user input “domain” and it can’t be resolved, the IP Phone will add and resolve the “domain” after user input.
Primary DNS	Main DNS server IP address.
Alter DNS	The second DNS server IP address.
PPPoE Configuration:	
Server	Service name, if PPPoE ISP has no special requirement for this name, generally use the default.
User	PPPoE account.
Password	PPPoE password.

Configure the parameter and then click "apply" to go into effect.

5.3.2 Local area network (LAN)

User can make local area network (LAN) configuration on this page, when bridging mode is selected, the local area network (LAN) configuration will no longer go into effect.



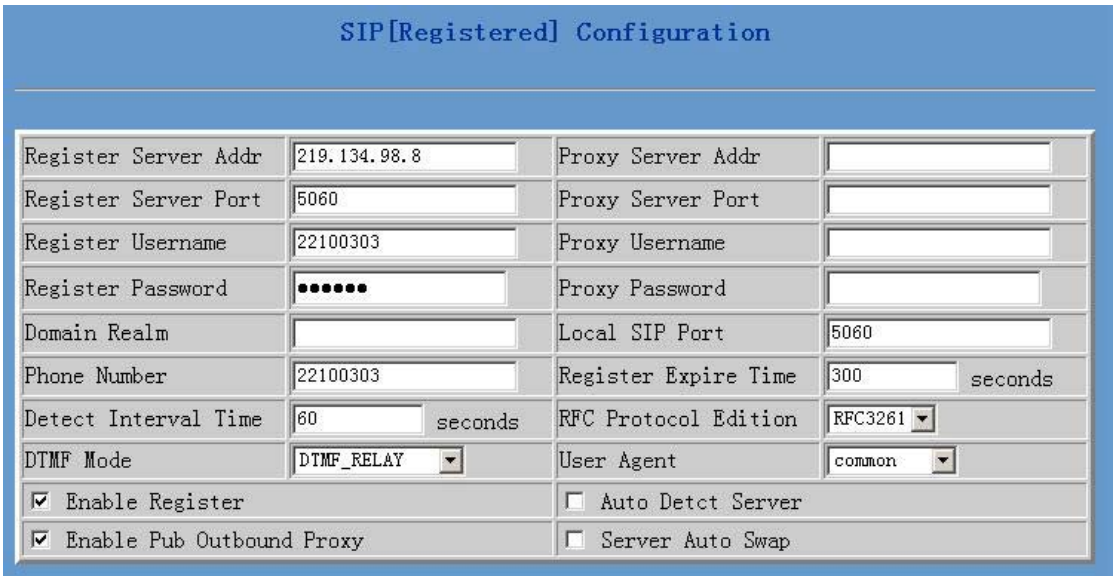
Configuration Explanation:

Bridge Mode	Use bridge mode (transparent mode): bridge mode will make the IP Phone no longer set IP address for LAN physical port, LAN and WAN will join in the same network.
IP	Configure LAN static IP.
Netmask	Configure LAN subnet mask.
DHCP Service	Enable LAN port DHCP server; after user modify LAN IP, the IP Phone will automatically modify the adjustment and save the configuration according to IP and subnet mask team DHCP Lease Table, user need to restart the IP Phone to make DHCP server configuration go into effect.
NAT	Enable NAT.

5.4 VOIP configuration

5.4.1 SIP configuration

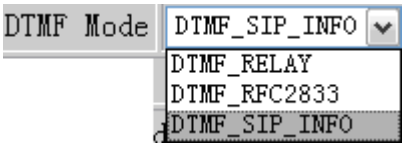
User can configure specific parameter of SIP signaling protocol on this page.



Configuration Explanation:

SIP[Registered] Configuration	Show SIP register state; if register successfully, there will show Registered in the square bracket, otherwise show Unregistered.
Register Server Addr	Configure SIP register server IP address.
Register Server Port	Configure SIP register server signal port.
Register Username	Configure SIP register account（usually it is the same with the port number that configured, some special SIP servers will have different port configurations, then the port configuration needs to be configured to be numbers, here the configuration account can be arbitrary character string）.

Register Password	Configure password of SIP register account.
Proxy Server Addr	Configure proxy server IP address（usually SIP will provide user with service of proxy server and register server which have the same configuration, so the configuration of proxy server is usually the same with that of register server, but if the configurations of them are different(such as different IP addresses), then each server's configuration should be modified separately.
Proxy Server Port	Configure SIP proxy server signal port.
Proxy Username	Configure proxy server account.
Proxy Password	Configure proxy server password.
Domain Realm	Configure SIP domain, which is provided by SIP Server. It can be the server IP (e.g. 202.112.10.12); or the domain name of the server(e.g. sipconcet.net); or other identifier provided by the server(e.g. sip).
Local SIP Port	Configure local signal port, the default is 5060（this port will go into effect immediately, the SIP call will use the modified port for communication after modification）.
Phone Number	Configure the phone number. If the accounts provided by ISP have no separate number, set the number to the same with server user name.

Register Expire Time	Configure expire time of SIP server register, the default is 600 seconds. If the expire time that server requires is more or less than that configured by the IP Phone, the IP Phone can automatically modify it to the recommended time limit and register.
Detect Interval Time	Configure detection interval time of the server, if the IP Phone enables SIP detection server function, the IP Phone will detect once for whether the server has response every other detection interval time.
RFC Protocol Edition	Enable the IP Phone to use protocol edition. When the IP Phone need to communicate with IP Phones which is using SIP1.0 such as CISCO5300 and so on, then it should be configured into RFC2543 to communicate normally. the default is to enable RFC3261.
DTMF Mode	DTMF sending mode configuration; three kinds: the above are basic configurations of SIP. 
User Agent	To be compatible with some special SIP Servers, the gateway has made some changes: Net2phone: server demands to use the MAC address as user agent; shengji: use special encrypt;
Enable Register	Configure enable/disable register.
Enable Pub Outbound Proxy	Configure to enable public outbound proxy. If proxy server has been enabled, the IP Phone will consider the user as using outbound proxy

	automatically. If the configuration has been disabled, the IP Phone can still be registered to the server, but can't make SIP call; configuration of registered call by the IP Phone will not have impacts on SIP point-to-point call.
Auto Detect Server	Configure automatic detection server of the IP Phone.
Server Auto Swap	Configure main and backup auto-swap server; if the IP Phone enables main and backup server function, the automatic detection and auto-swap functions should both be chosen.

Note: if you want to register and call through server, you must configure corresponding numbers（which are usually SIP accounts） to local port, otherwise the IP Phone will reject for sending out register message when it considers that there is no number.

After the aforesaid network and VoIP configuration, if the network can communicate correctly, the user should be able to make VOIP calls through the register and proxy.

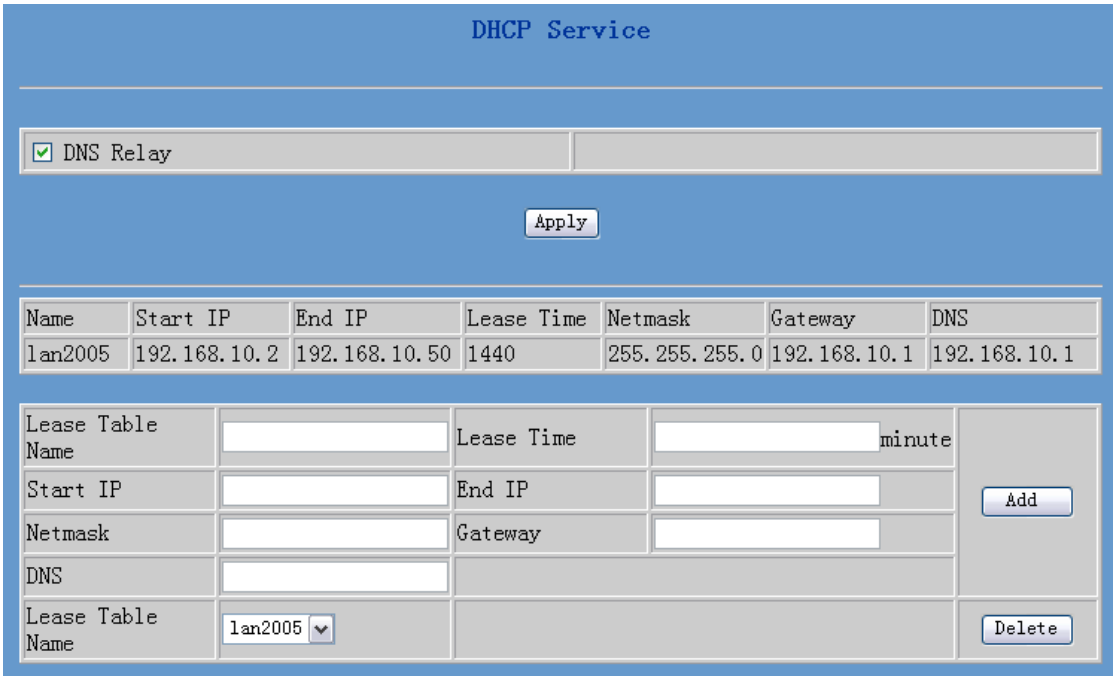
After submitting the configuration, if the status is still incorrect, click the menu in the left to renovate. Please do not click the “apply” continuously, since some ISP may inhibit the IP Phone to register or cancel continuously. If users click “apply” continuously, or register and cancel frequently, the server may stop response, this will make the IP Phone can not receive the certifications of register/cancel request and show error register status!

5.5 Advance configuration

4.5.1 DHCP configuration

User can set up the DHCP service, customer can from the definition

dynamic state IP allotment scope and other install etc. on this page.



Configuration Explanation:

DNS Relay	Configure DNS Relay method; This way can use an equipments of conjunction LAN to use LAN IP directly Be the DNS server address, recognize tacitly to open. Click apply to take effect after pick out;
Lease Table Name	Configure Lease Table Name
Lease Time	Configure Lease Table of increase leases IP time limit
Start IP	Configure the start IP address of the Lease Table of increase: LAN allotment address of the time starts search an idle IP address to divide from this address to at LAN the equipments that use DHCP.
End IP	Configure the end IP address of the Lease Table of increase: Start to end an address of the piece come to a decision to connect can get into the LAN

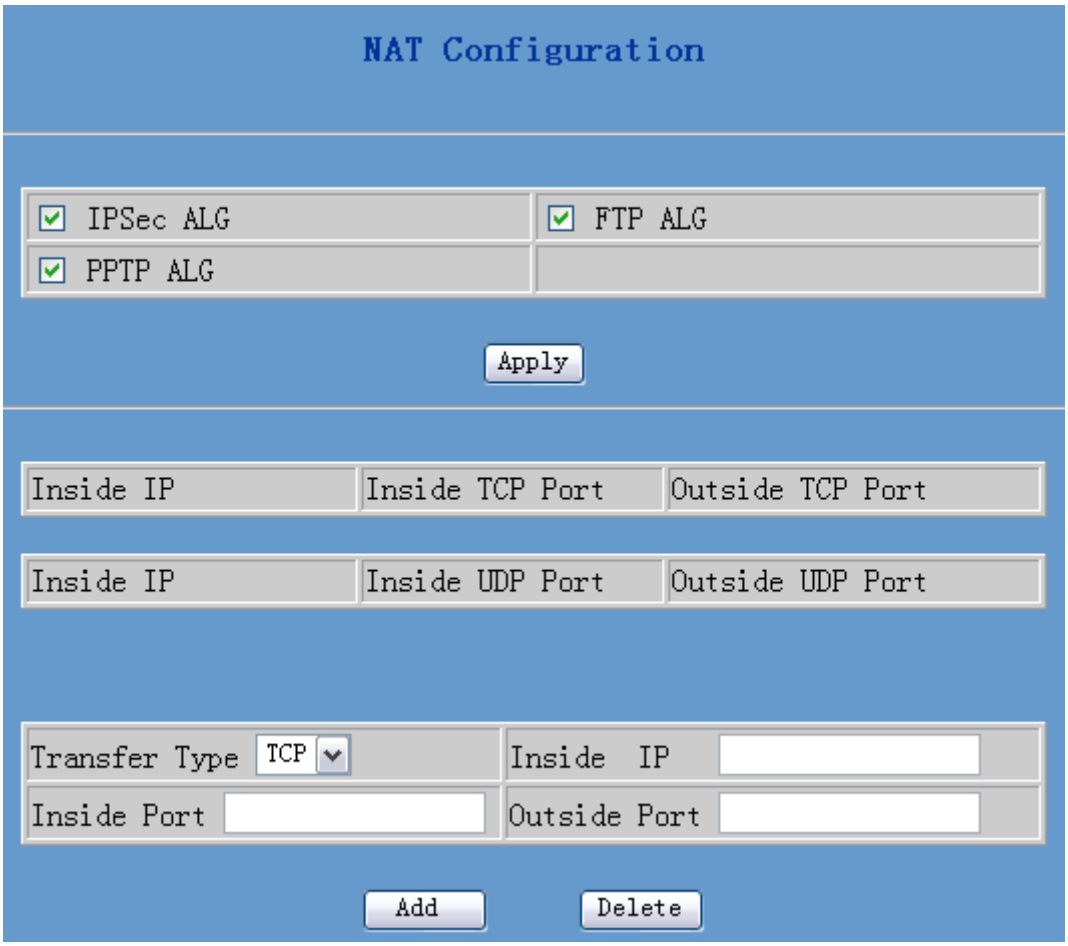
	oscular network equipments of IP number. A connects the network equipments DHCP to gain into the LAN to of the address lies between the Start IP and the End IP necessarily;
Netmask	Configure the netmask of the Lease Table of increase.
Gateway	Configure the gateway of the Lease Table of increase.
DNS	Configure the DNS of the Lease Table of increase.
After the configuration has been done, click “Add” to make them go into effect.	
Lease Table Name	The choice descends to pull menu in of want to delete of the Lease Table Name, click the key of Delete to delete the options in the DHCP Lease Table.

Note: If customer modify lease Table, needing to save and reboot.

4.5.2 Nat configuration

User can set up IPSec ALG, FTP ALG, PPTP ALG and NAT on this page.

Can install 10s at most.



Configuration Explanation:

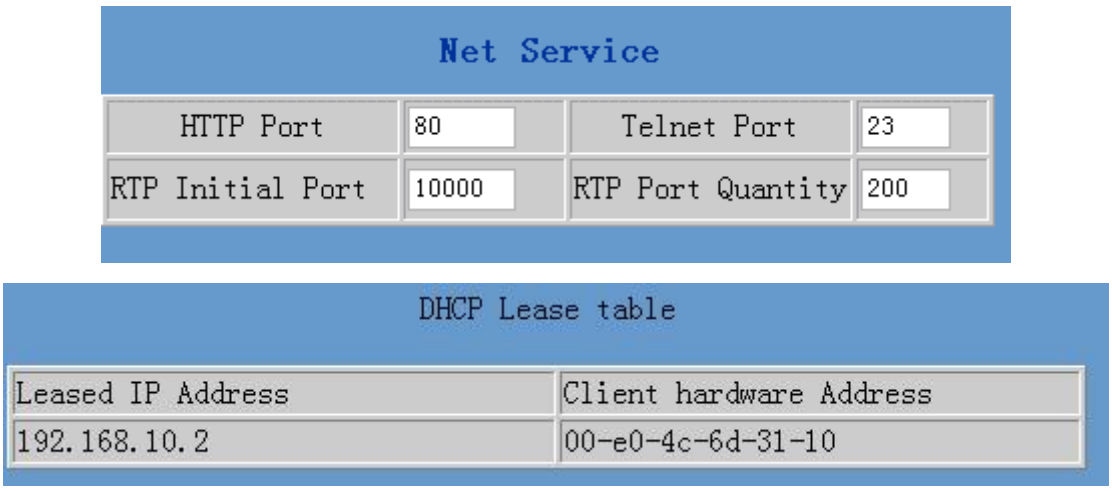
IPSec ALG	Configure IPSec ALG and the tacit approval open.
FTP ALG	Configure FTP ALG and the tacit approval open.
PPTP ALG	Configure PPTP ALG and the tacit approval open.
Transfer Type	Configure Transfer Type: TCO or UDP.
Inside IP	Configure LAN equipments IP address.
Inside Port	Configure LAN equipments port.
Outside Port	Configure Outside Port.

After the configuration has been done, click “Add” to make them go into

effect.

4.5.3 Net Service configuration

User can set up Telnet, HTTP, RTP port on this page and view DHCP table.



Configuration Explanation:

HTTP Port	Configure web browse port, the default is 80 port, if you want to enhance system safety, you'd better change it into non-80 standard port.
Telnet Port	Configure telnet port, the default is 23 port.
RTP Initial Port	Enable RTP initial port configuration. It is dynamic allocation.
RTP Port Quality	Configure the maximum quantity of RTP port. The default is 200.
Leased IP Address	Leased IP-MAC correspondence table of DHCP.

Note:

1) The configuration on this page needs to be saved after modified and will go into effect after restarting.

2) If the Telnet, HTTP port will be modified, the port is better to be set as greater than 1024,because the 1024 port system will save ports.

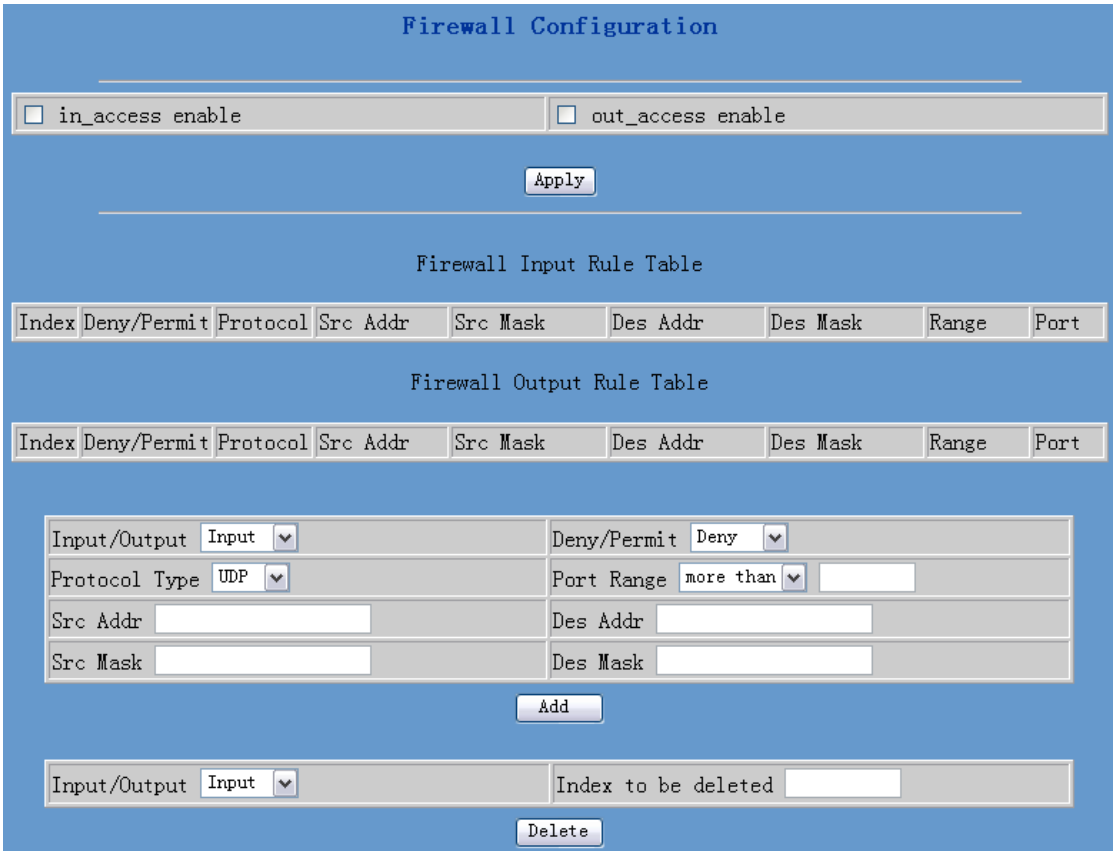
3) Set the HTTP port as 0, then the http service will be disabled.

4.5.4 Firewall configuration

One this page user can establish whether be in use an input and output firewall, can install the rule of input and output, make use of these install and can keep some malevolent IPs come to visit to ask this ATA, or the restriction visit an outside net of some resourceses, raise a safety.

This function supports two kinds of rules: Input rule and output rule. The regulation rule then will assign an ordinal number each time. Biggest allow each rule each install 10s.

This function is according to the percolation of the equipments WAN, namely the source address of the output and the purpose address of the input should be a WAN IP.



Configuration Explanation:

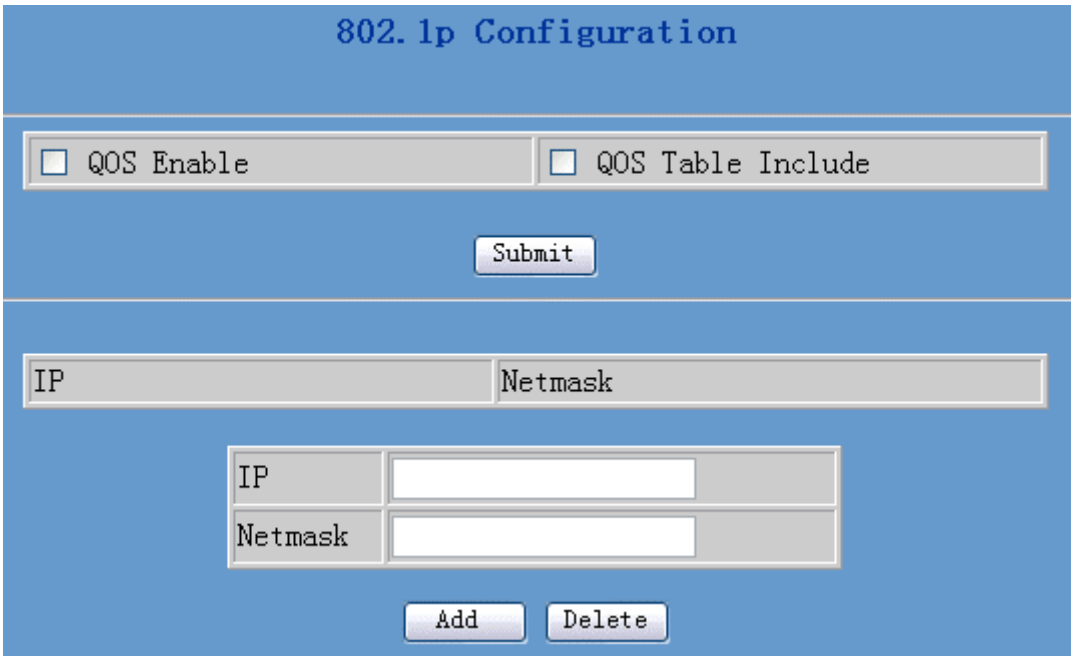
in_access enable	Be in use the input rule application
out_access enable	Be in use the output rule application
Input/Output	The choice increases rule at present is an input or output rule
Deny/Permit	Choice the current rule install is forbid or allow
Protocol Type	Configure protocol type
Port Range	Configure port range
Src Addr	Configure source address. The source address can be a concrete IP address, can be also a network address.
Src Mask	The netmask of the source address, be to install to explain to be a concrete host for 255.255.255.255. be

	explain to be a concrete host for 255.255.255.255, be to install to explain to be a network ID for 255.255.255.0.
Des Addr	Configure destination address. The destination address can be a concrete IP address, can be also a network address.
Des Mask	The netmask of the destination address, be to install to explain to be a concrete host for 255.255.255.255, be to install to explain to be a network ID for 255.255.255.0.

After the configuration has been done, click “Add” to make them go into effect.

4.5.5 QOS configuration

What ATA carry out is according to the QOS with 802.1 p, used for in the data network/MAC layer marking network correspondence and line up to settle a network to correspond by letter of have the initiative class. The 802.1 p correspondence will be categorize and deliver to the destination.



Configure explanation:

QOS Enable	Install to be in use/forbid to use a QOS service
QOS Table Include	Be in use the QOS Table Include, mean that set now of QOS table inside of the net segment address is the address that needs to provide the QOS service, the address out of the table doesn't provide QOS; Don't be in use QOS Table Include, explain all addresses out of the table need to provide the QOS service.
IP	Configure the IP address of the Qos service.
Netmask	Configure the netmask address of the Qos service.

Note: The IP constitution can be the network address, can be also a concrete IP. Can pass the constitution of the IP and netmask to carry on enactment to the purpose address. Set for 255.255.255.255, then is appointed concrete IP.

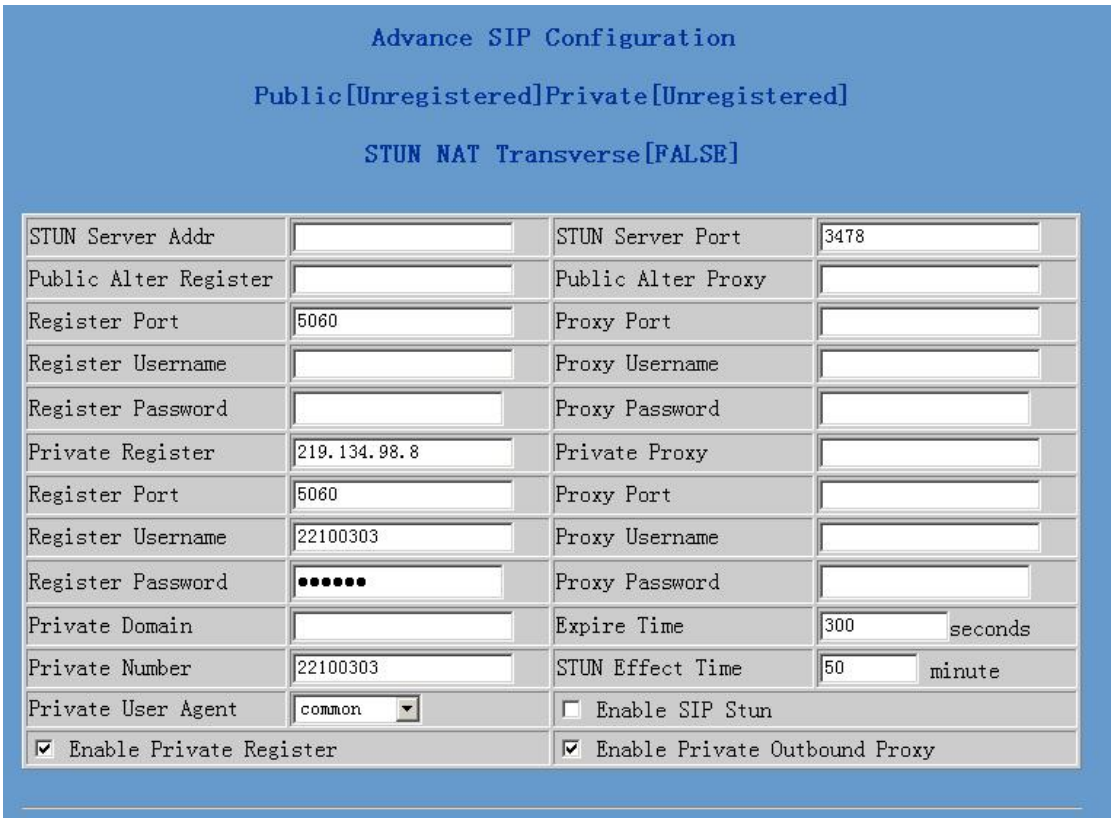
4.5.6 SIP advanced configuration

Set SIP STUN, private and backup server, user password and so on.

SIP STUN is a kind of server that used to realize the SIP's enablement of NAT, when the STUN server IP of the gateway has been configured （generally the default is 3478）and Enable SIP Stun has been selected, conventional SIP server can be used to realize the gateway's penetration of NAT.

Public backup server can implement the proxy of the dialogue machine through auto-swap function when no response to public server. When the gateway detect response of public server, it will auto-swap to public server. Public backup server is redundancy backup of public server, it should have the same account with public server.

The gateway's supports two different kinds of SIP server synchronously，In this way user can register and use two different kinds of service synchronously.



Configure explanation:

1) Configure of private server:

Public [unregistered] Private [Unregistered]	To show whether the gateway has been registered on public server or private server.
STUN Server Addr	Configure IP address of SIP STUN server.
STUN Server Port	Configure port of SIP STUN.
	STUN can support SIP terminal's penetration to NAT in the inner-net. In this way, as long as there is conventional SIP proxy and a STUN server placed in the public net, it will do; but STUN only supports three NAT modes: FULL CONE, restricted, port restricted.

2) Public backup server configuration:

Public Alter Register	10.1.1.11	Public Alter Proxy	0.0.0.0
Register Port	5060	Proxy Port	5060
Register Username	1234	Proxy Username	1234
Register Password	••••	Proxy Password	••••

The specific configuration parameter has the same meaning with public server. It should be noted that the username and password should be the same with the public main server;

3) Private server configuration.

Private Register	210.25.132.124	Private Proxy	210.25.132.124
Register Port	5060	Proxy Port	5060
Register Username		Proxy Username	
Register Password		Proxy Password	

Specific configuration parameter has the same meaning with public server;

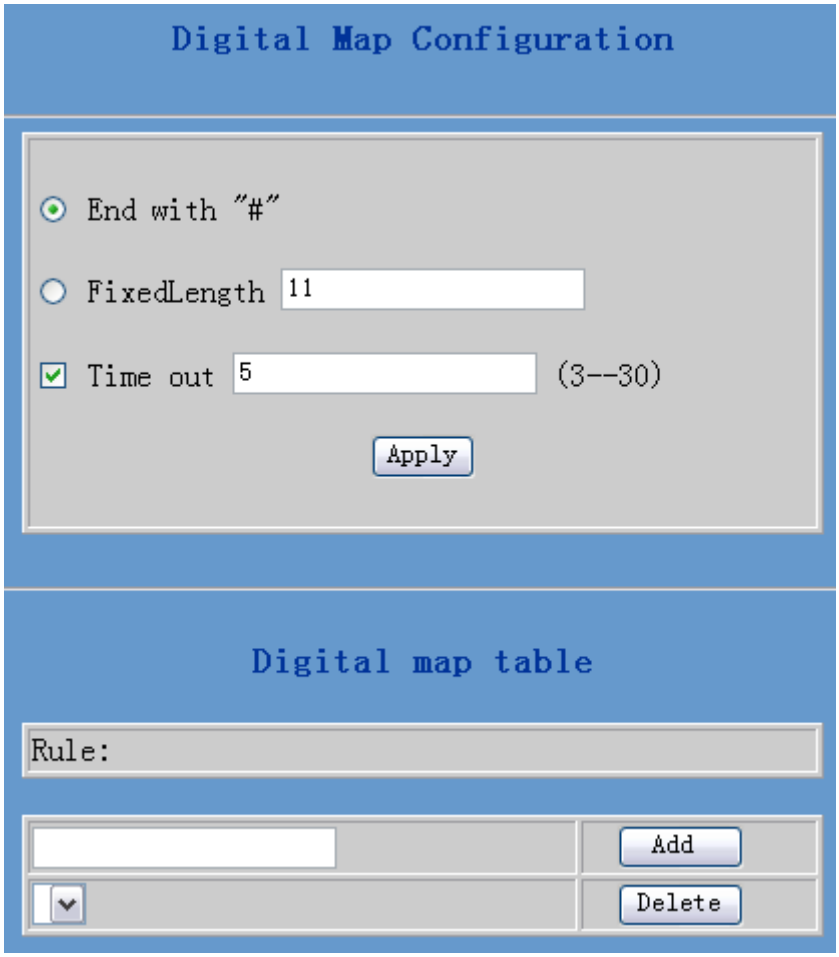
4) Others

Private Domain	Configure SIP Domain of SIP private server.
Private Number	The port number of private server. When no special requirement, it is the same with user name of the server.
Private User Agent	Configurate User Agent of private server, the default is common.
Expire Time	Configurate the register expire time of private server.
STUN Effect Time	Interval time for STUN's detection on NAT type, the unit is minute.
Enable SIP Stun	Configure enable/disable SIP STUN.
Enable Private Register	Configure permit/deny private server register.
Enable Private Outbound Proxy	Configure enable/disable private outbound proxy.

After the aforesaid configuration has been done, click "apply" to make them go into effect.

4.5.7 Digital Map configuration

On this page, user can set up call service.



Configuration Explanation:

End with "#"	With # the number end: Added finally when customer make a phone call # number.
FixedLength	Accept number by fixed-length: The number that the system is input with the fixed length interception customer.
Time out	Use super hour accept number: Super the system sends out the number accept after hour.

Rules:
[1-8]xxx
9xxxxxxx
911
99T4
9911x.T4

The rule explain:

[] []Is an appointed digital scope. Can be a scope, can also be stir number to partition, can be also the row form of digital.

X The x is to match arbitrarily.

. BE match arbitrarily length, include empty.

Tn The Tn means to accept the empress of the number to end after n second. The n is compulsive, scope is 0-9. The Tn musts be the lasts to install. Lack province unqualified place Tn words, the system think to be a T0, accepting number be over immediately namely.

Install for instance:

[1-8] xxx, mean from 1000-8999 of havings 4 number for grows send out immediately after accepting together 4.

9xxxxxxx, mean with 9 number of the beginnings, send out immediately after accept together total 8 numbers.

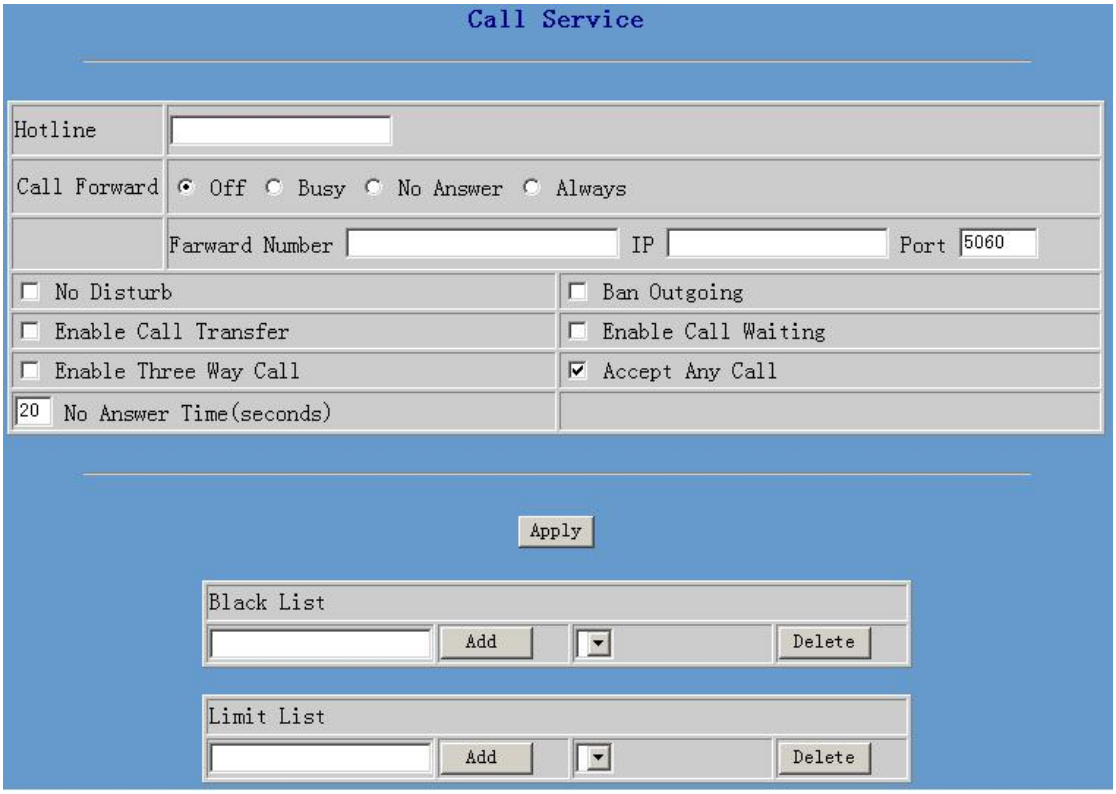
911, mean 911 this numbers, send out immediately after stiring to finish.

99T4, mean 99 this numbers send out after stiring 4 through.

9911x.T4, mean with 9911 beginnings of, at least 5 numbers, Be receiving future reunion to send out after 4.

4.5.8 Call service configuration

On this page, user can set value-added services such as hotline, call forwarding, call transfer (CT), call-waiting service, three way call, blacklist, out-limit list and so on.



Configuration Explanation:

Hotline	Configure hot-line number of the port. With this number of the port, this hot-line number will be dialed automatically as soon as off-hook and user can't dial any other number.
Call Forward	Call forwarding. The default is Disable. when busy is selected, then when the gateway is engaged, it will automatically transfer to the configured number according to the following configuration; when always is selected, then the gateway will directly transfer all the numbers that dial to this port to the configured numbers.
Forward Number/ IP/Port	Number, IP, Port configuration of call forward （CWD）.

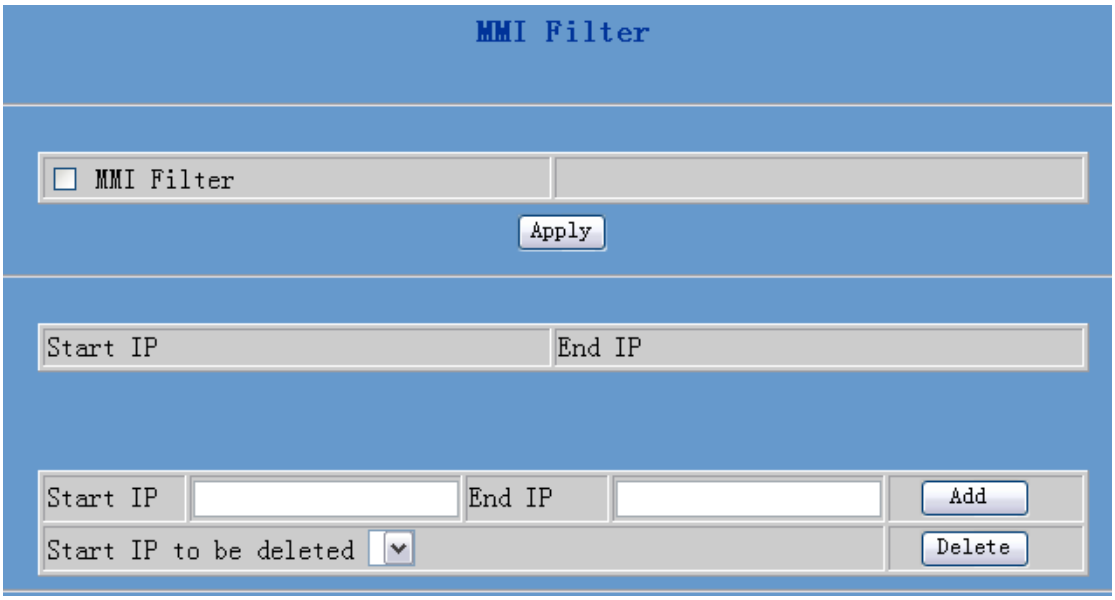
Enable Call Waiting	Configure enable/disable call waiting service. After it is enabled, user can hold calls of the other party by hooking, with hooking again, the hold call can go on.
Enable Call Transfer	Configure enable/disable call transfer (CT). After it is enabled, user accept calls, with hooking and dial directly, the gateway will transfer the calls according to the above configurations of the port number IP images.
Enable Three Way Call	Configure enable/disable three way call. user can call the other part as the call origination, after talking, make hooking to hold this part and then press * key to hear the dialing tone, after call completion to the third party, hooking again to recover the talk with the second part, then the three way call synchronously.
After the aforesaid configuration has been done, click “apply” to make them go into effect.	
Black List	Configure add/delete blacklist. If user don't want to answer a certain number, please add this number to the list, and then this number will be unable to get through the gateway.
Limit List	Configure out-limit list. for example, if user don't want the gateway to dial a certain number, please add the number to this table, and the user will be unable to get through this number.

4.5.9 MMI Filter configuration

MMI: the Man Machine Interface.

On this page customer can establish to allow one MMI of the machine

interview ATA of the some IP address segment to install management words machine.

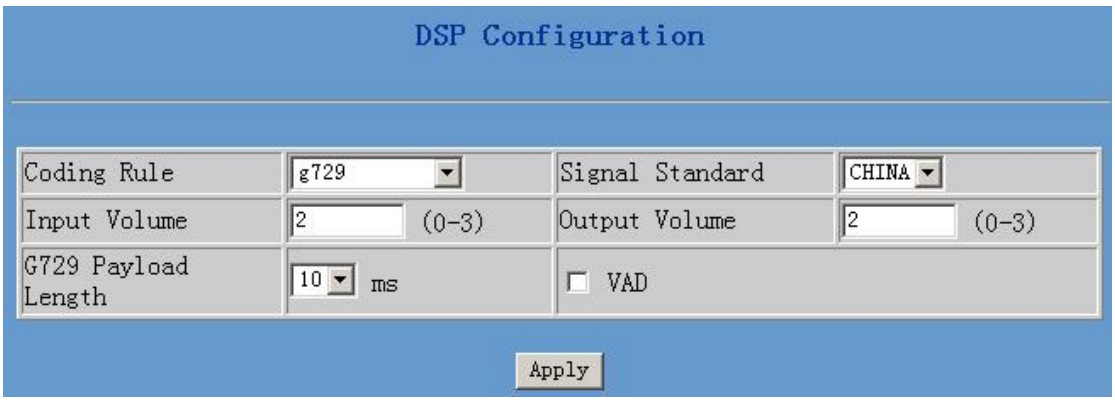


Configuration Explanation:

MMI Filter	Install to be in use/forbid to filter with the MMI interview
Start IP	Configure Start IP.
End IP	Configure End IP.

4.5.10 DSP configuration

On this page, user can set speech coding, I/O volume control, cue tone standard, caller ID standard and so on.



Configuration Explanation:

Coding Rule	Configure DSP Speech coding, choose the method to process.
Signal Standard	Configure the signal tone standard. For example, dial tone, ringback tone, busy tone and so on. It contains standards of China, United States, German, etc.
Input Volume	Configure input volume.
Output Volume	Configure output volume.
G729 Payload Length	Configure G729 Payload Length according to the requirement.
VAD	Voice Activity Detection. When the telephone has no input voice, for saving bandwidth, use VAD to inform DSP to transfer silent frames.

After the configuration has been done, click “apply” to make them go into effect.

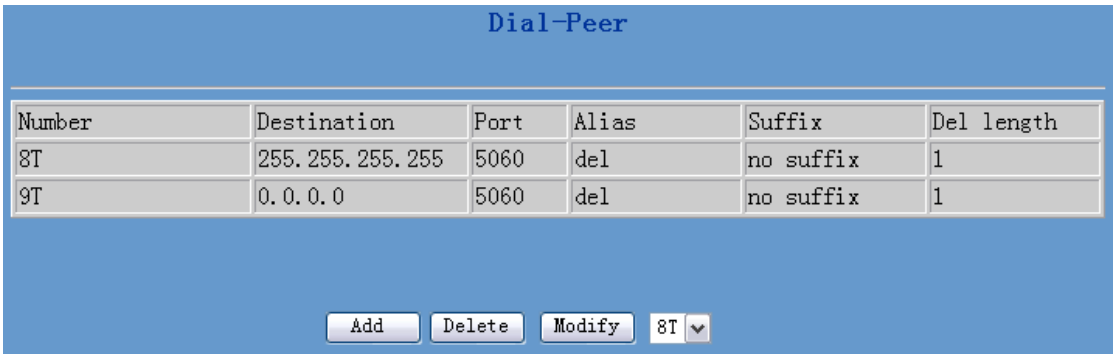
5.6 Dial Peer configuration

Number IP table configuration:

Function of number IP table is one way to implement the IP Phone's calling online, and the calling of the IP Phone will be more flexible by configurating the number IP table. For example, user know the other party's number and IP and want to make direct call to the party by point-to-point mode: the other party's number is 1234, make a configuration of 1234 directly ,then the IP Phone will send the called number1234 to the corresponding IP address; Or set numbers with prefix matching pattern, for example, user want to make a call to a number in a certain region（010）, user can configure the corresponding number IP as 010T--protocol--IP, after that, whenever user dial numbers with 010 prefix（ such as 010－62201234), the call will be made by this rule.

Bases on this configuration, we can also make the IP Phone use different accounts and run speed calling without swap.

When making deletion or modification, select the number first, then click Modify and complete the operation.



Configuration explanation and basic application of the number IP table:

1) The above table displays calling number IP image list. The table shows different configuration cases when use different protocols.

- ✓ Outgoing call through lifeline, dial an extra “*”;
- ✓ To call directly through point-to-point, the number is 166;
- ✓ For the terminal with ip 192.168.0.166, dial an extra “9” when call

through private SIP server.

By configure different prefixes, protocols and IP formats, a outgoing call can be made by different accounts or platforms.

2) Click “Add”, a figure will be shown at the lower part of the page, of which:

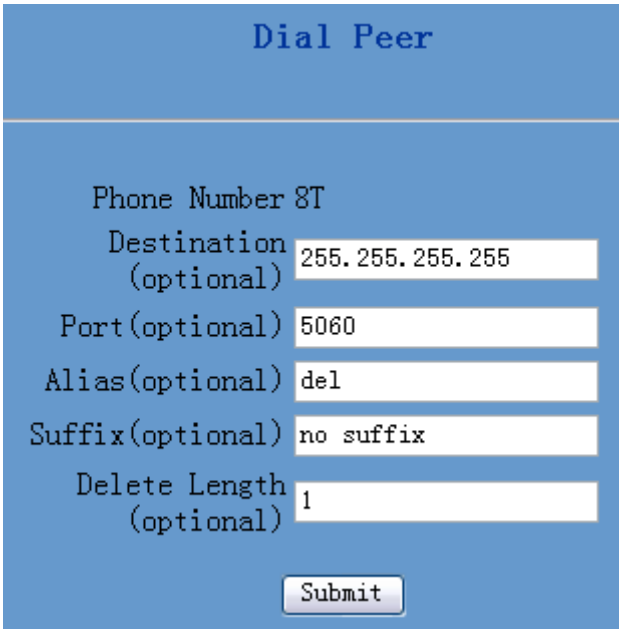
Phone Number	It is to add outgoing call number, there are two kinds of outgoing call number setup: One is exactitude matching, after this configuration has been done, when the number is totally the same with the user’s calling number, the IP Phone will make the call with this number's IP address image or configuration; Another is prefix matching（be equivalent to PSTN’s district number prefix function）, if the previous N bits of this number are the same with that of the user's calling number (the prefix number length), then the IP Phone will use this number's IP address image or configuration to make the call. When configurating the prefix matching, letter "T" should be added behind the prefix number to be distinguished from the exactitude matching; the longest length is 30 bits.
Destination	Configure destination address, if it is point-to-point call, then input the opposite terminal's IP address, it can also be set as domain name and resolved the specific IP address by DNS server of the IP Phone. If no configuration has been made, then the IP will be considered as 0.0.0.0. This is an optional configuration item;
Port	Configure the other party's protocol signal port, this is optional configuration item: when nothing is input, then the default of h323 protocol is 1720, the default

	of sip protocol is 5060; lifeline required no configuration of this item, shown as 0;
Alias	Configure alias, this is optional configuration item: it is the number to be used when the other party's number has prefix; when no configuration has been made, shown as no alias; alias can be divided into four types, it should be combined with replacing length to make the setup: Add: xxx, add xxx before number. in this way it can help user save the dialing length; All: xxx, the number is all replaced by xxx; speed dialing can be implemented, for example, user configure the dialing number as 1, with the configuration "all" , the actual calling number will be replaced; Del, delete n bit in the front part of the number can be decided by the replacing length; this configuration can decide the protocol for appointed number; Rep: xxx, n bit in the front part of the number will be replaced. n is decided by the replacing length. For example, user want to dial PSTN（010－62281493）by VoIP's voice over service, while actually the called number should be 8610－62281493, then we can configure called number as 010T,then rep: 8610, and then set the replacing length as 3. So that when user make a call with 010 prefix, the number will be replaced as 8610 plus the number and then sent out. It is a convenient thinking mode for user to make a call.

Suffix	Configure suffix, this is optional configuration item: it is the additive dial-out number behind the number; when no configuration has been made, shown as no suffix.
Delete Length (optional)	Configure the replacing length, replace the number that user input according to this length; this is optional configuration item.

3) Click “Delete”: Delete selective number IP image;

4) Click “Modify”: Choose a corresponding phone number image in the drop-down menu and then click “modify” to make modification. Then the page changes to the following chart. After modification, click “Submit” to go into effect.



Phone Number	This is the selected number(to be modified), read-only;
Destination (optional)	To modify destination address, this is optional configuration item;

Port(optional)	To modify destination IP Phone port; this is optional configuration item;
Alias(optional)	To modify alias, this is optional configuration item;
Suffix(optional)	To modify suffix; this is optional configuration item;
Delete Length (optional)	To modify replacing length（if rep and del of alias have been configured）

Click submit to go into effect.

How to configure IP table of number to implement configuration of using multi-accounts synchronously**:**

For example, now user has two SIP accounts, then under the default condition, user can only make calls by the public SIP server. Configure the number IP table to select the call protocol, and user could select different accounts to make calls freely.

The configuration process will not be repeated, now I will mainly introduce what kind of number IP image can implement this function.

By configuration, image table as follows will be gained:

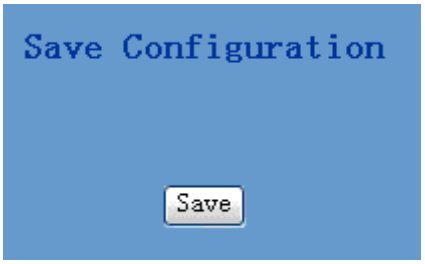
Number	Destination	Port	Alias	Suffix	Del length
9T	0.0.0.0	5060	del	no suffix	1
8T	255.255.255.255	5060	del	no suffix	1

Image of 9T means when user configure public SIP server and register, then user just need to add a “9” before the calling number whenever making a call by public SIP;

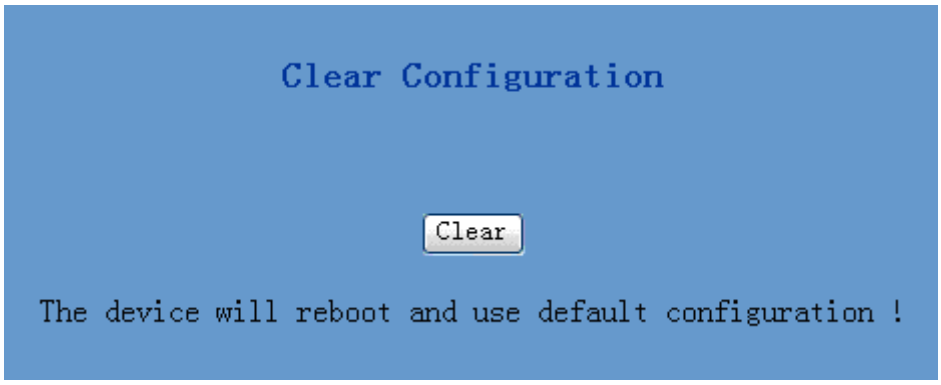
Image of 8T means when user configure private SIP server and register, then user just need to add a “8” before the calling number whenever making a call by private SIP;

5.7 Save and Clear configuration

User can save the current configuration on this page.



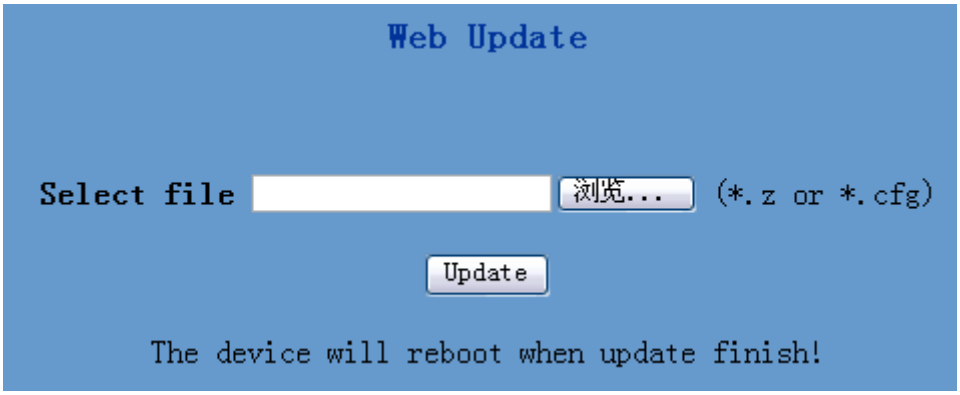
The system configuration can be set as factory default configuration on clear configuration page and the IP Phone will restart automatically. Only user who is administrator could clear configuration as factory default configuration. Guest can only clear configuration excluding SIP.



5.8 Upgrade on-line

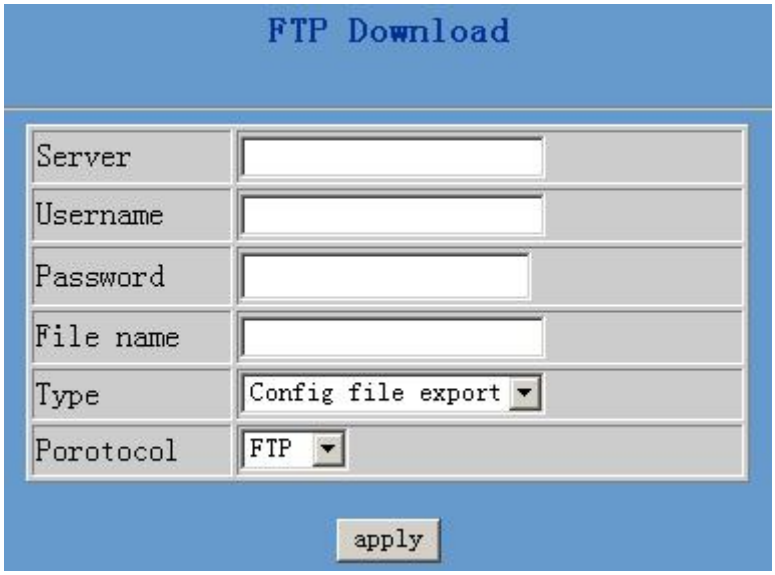
5.8.1 Upload WEB page

On this page, user can select the upgrade documents (**firmware or config file**) on hard disk of the computer directly to run the system upgrade. After the upgrade has been completed, restart the IP Phone and it will be usable at once.



5.8.2 FTP download

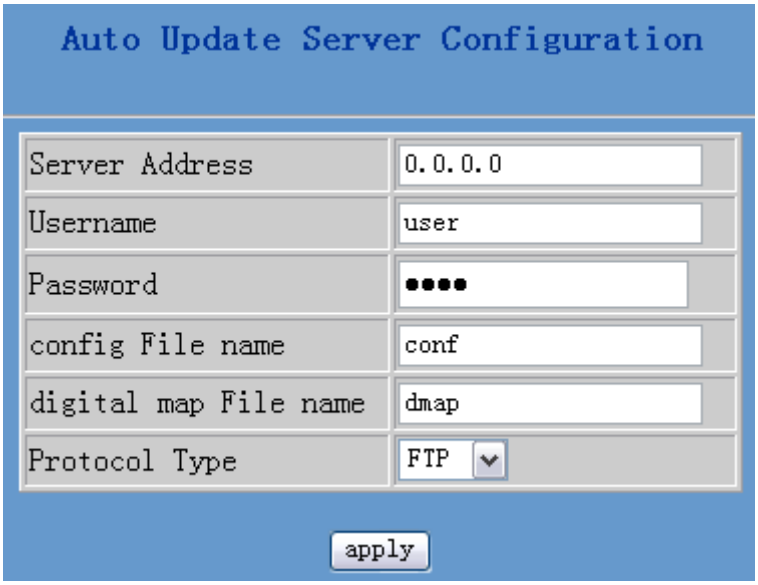
On this page, user can upgrade system and configure files by FTP or TFTP mode.



Configuration Explanation:

Server	Configure upload or download FTP/ TFTP server IP address;
Username	Configure username of the upload or download FTP server. If user select TFTP mode, username and password are not required to be configured;
Password	Configure upload or download of FTP server password;
File name	Configure upload or download system upgrade document or system layout file name. It should be noted that system file take .dlf as suffix, configuration files take .cfg as suffix;
Type	Configure the operation type. There are 3 types: ✓ "application update": update the system through FTP or TFTP;

	✓ “config file export”: Upload the IP Phone configuration file to the remote server through ftp or TFTP protocols; ✓ “config file import”: Download the configuration file from remote server and import to local IP Phone
Protocol	Select server type;



Configuration Explanation:

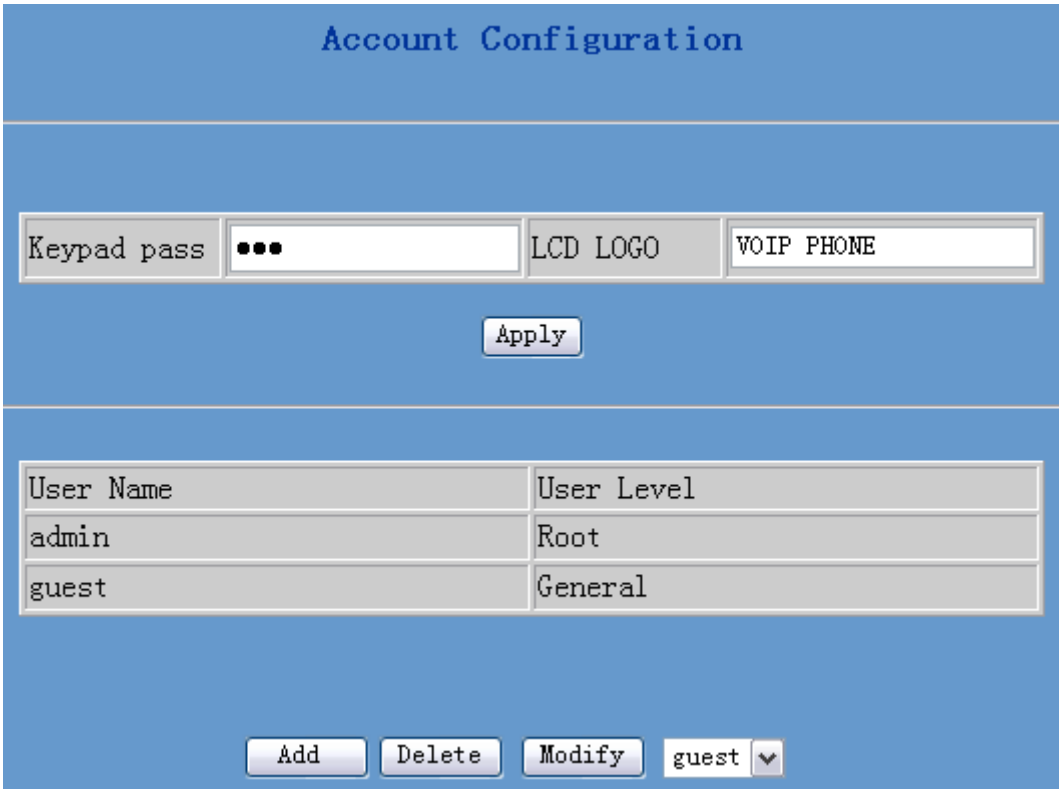
Server Address	Configure update or download FTP/ TFTP server IP address;
Username	Configure username of the Auto update FTP server. If user select TFTP mode, username and password are not required to be configured;
Password	Configure Auto update of FTP server password;
config File name	Configure name of config file;
digital map File name	Configure map File name

Protocol Type	Select Protocol Type: FTP and TFTP;

5.9 System management

5.9.1 Account management

On this page, user can add and delete users according to own needs and can modify user's authorities there have been.

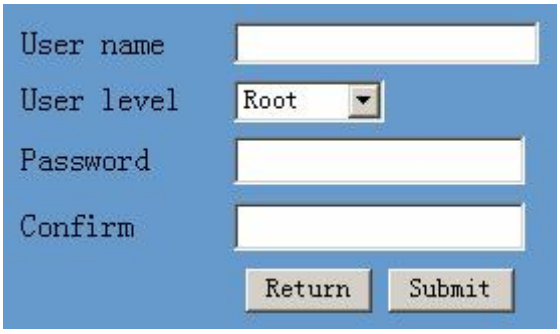


Configuration Explanation:

Keypad pass	Configure password of the keypad;
LCD LOGO	Configure LCD LOGO, this LOGO will show when ATA reboot;

1) The table in the page displays the IP Phone user account list;

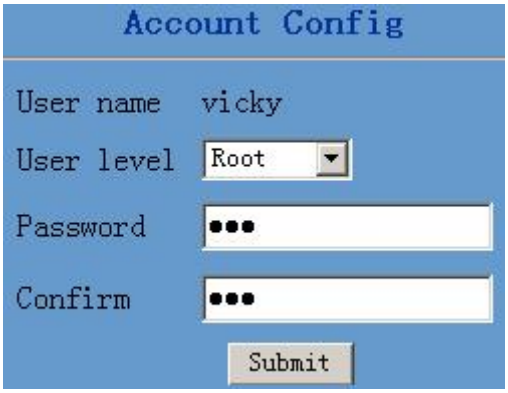
2) Click “Add” to add ATA account; it will be shown at lower part of page as the following figure, of which:



User Level	As account level; root possesses authorities to modify configuration, general possesses read-only authority;
Password	As corresponding password of the additive account;
Confirm	As second confirmation of password, to ensure correct setup of password;
Return/Submit	Click “Submit” to go into effect; click “Return” to cancel configuration.

3) Click “Delete” to delete the selected account from the drop-down menu.

4) Click “Modify” to modify the chosen account, need to select account first, click load again and then click modify, it will be shown at lower part of page as the following figure, of which:



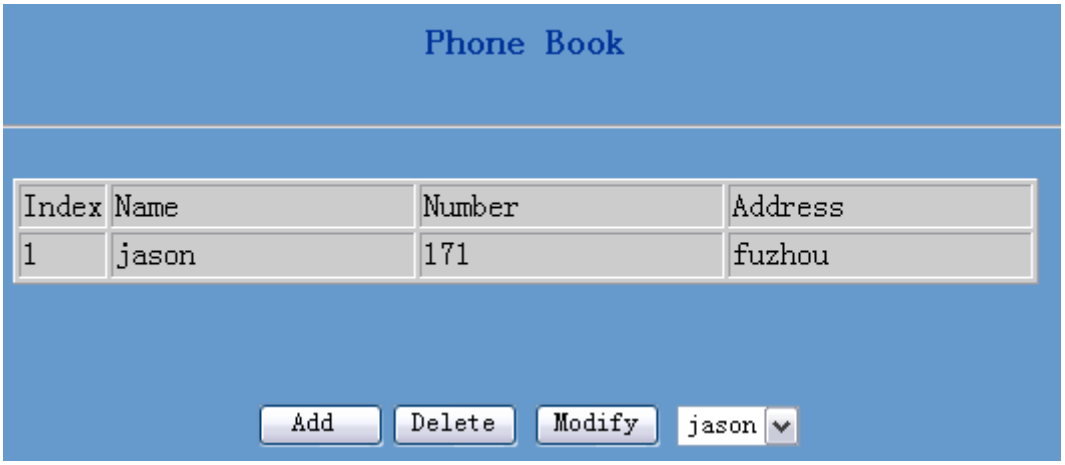
User name	The modified username;
User level	Modify user authorities;
Password	Modify user password;
Confirm	Make confirmation of the modified user password;
Submit	Click “Submit” to submit the modification .

Owing to the phone's default account: accounts of the administrator level-admin and the ordinary level—guest are all weak account and weak password, the username and password will be easily to be guessed on public network, so the user had better modify the administrator and ordinary user.

Enter with manager level when making modification，create a administrator account and a browse account（you'd better not set the name as admin, administrator, guest, etc.）， set password and then save configuration，entering with new manager account, delete default manager and browse account and save configuration，security will be enhanced!

5.9.2 Phone Book

On this page, user can add and delete phone book according to they needs and can modify phone book.

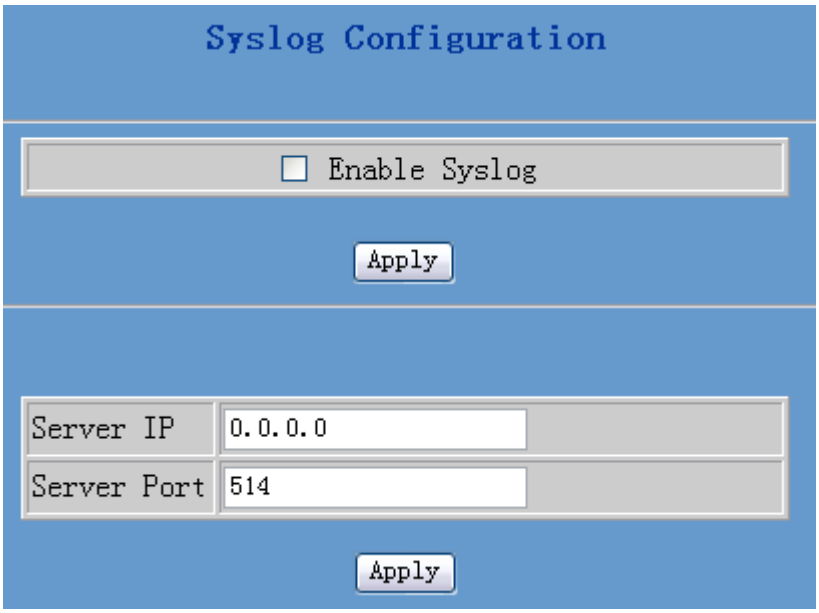


Configuration Explanation:

Index	serial number;
Name	Configure name of phone book or man;
Number	Configure phone number
Address	Configure home address

5.9.3 Syslog Config

On this page, user can set Syslog Server.

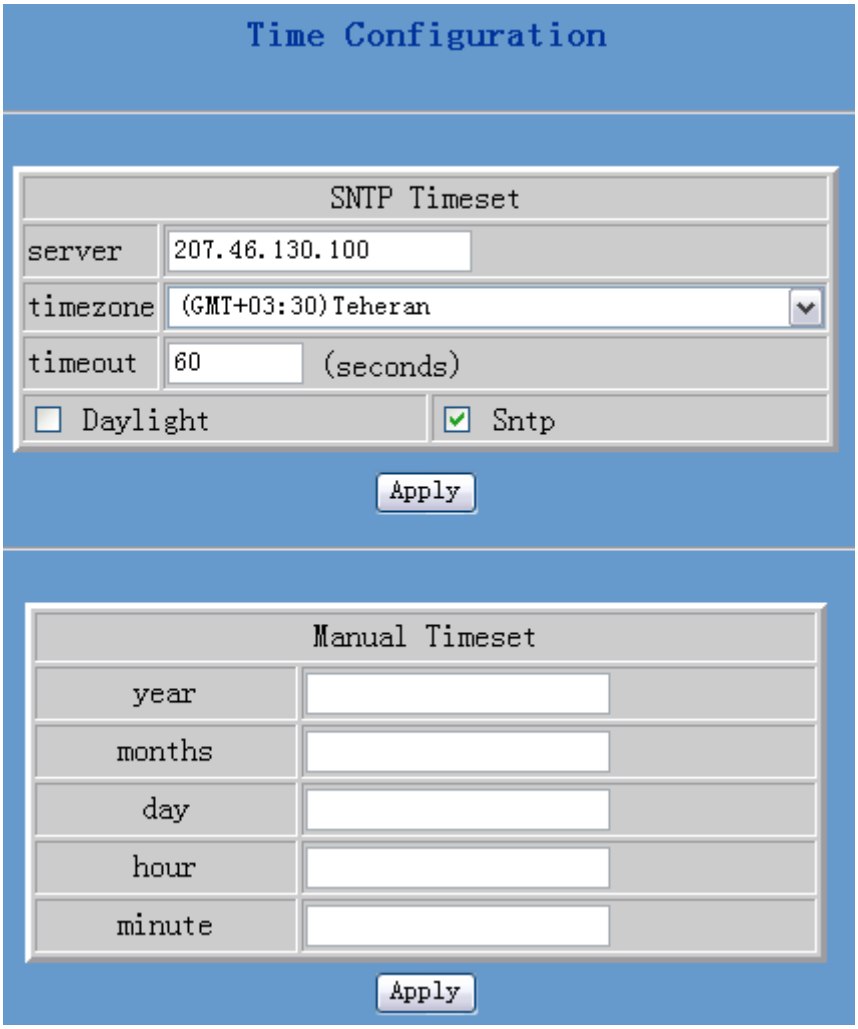


Configuration Explanation:

Enable Syslog	Configure enable/disable Syslog ;
Server IP	Configure syslog Server IP;
Server Port	Configure syslog Server Port

5.9.4 Time Set

On this page, user can set time.



Configuration Explanation:

server	Configure SNTP server IP adderss ;
timezone	Configure timezone;
timeout	Configure timeout
Daylight	Configure enable/disable daylight
Sntp	Configure enable/disable sntp timeset;

5.9.5 Reboot

On this page, user can reboot IP Phone.

Reserves the right to make changes in technical and product specification without prior notice.

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IPT1000H MANAGER MANUAL

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